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Improvement of wiener filter based speech enhancement using compressive sensing (Conference Paper)

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Abstract

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Many researches have been addressed on design approach for speech enhancement. They are mainly focus on speech quality and intelligibility to produce high performance level of speech signal. Wiener filter is one of the adaptive filter algorithms to adjust filter coefficients and produce an output signal that satisfies some statistical criterion. The objective measures will optimize using informal listening test and Perceptual Evaluation of Speech Quality (PESQ). The cascaded design approach of the Wiener filter and compressive sensing (CS) algorithm with random matrices were applied to exhibit and produce the better results. Therefore, applying the speech signal to this algorithm design in terms of appropriate basis functions of relatively few nonzero coefficients in CS can achieve an optimal estimate of uncorrelated components of noisy speech without obvious degradation of speech quality. Aside from that, this algorithm can be promised the speech enhancement with high performance results and significantly improved comparing to classical methods. © 2014 IEEE.

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